AD-A031 920 TRACOR INC AUSTIN TEX F/G 17/1
A GLOSSARY OF SONAR TECHNOLOGY TERMS.(U)
DEC 71 N00024-69-C-1083
UNCLASSIFIED T71-AU-9616-U NL

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24 Jan 72

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Contract N00024-69-C-1083 Add
TRACOR Document No. T71-AU-9616-U

A GLOSSARY OF SONAR TECHNOLOGY TERMS

Submitted to

Commander
Naval Ship Systems Command
Department of the Navy
Washington, D. C. 20360

Attention: Code 901

13 December 1971

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TRACOR Document No. T71-AU-9616-U

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FOREWORD

This glossary of terms was compiled for use as a ready reference for persons technically trained but relatively new to sonar technology. As such this compilation was prepared under the assumption that users have a working knowledge of engineering physics and mathematics. No attempt was made to produce a fundamental acoustics primer, since several standard texts exist. Classified terms were excluded in order to allow this glossary to be freely used as a desk top reference.

The definitions presented here are consistent with concepts in use by the sonar community, both within the Navy sonar offices and industry. These definitions are consistent with the "standard" definitions for certain acoustic terms such as available from the American Standards Association, Inc., and the American Institute of Physics, although no effort was made to utilize the definitions of those groups.

Users will appreciate that the entires in this glossary appear in alphabetical order. A Table of Contents lists all definitions provided and is useful both in locating the pages of interest, and in determining whether a particular term is included. Certain terms used in definitions are themselves the subject of a definition elsewhere in this glossary. These terms appear in all capital letters as a cross-reference clue to the interested reader.

This glossary is one of many results produced under Contract N00024-69-C-1083 for NAVSHIPS Code 901. TRACOR acknowledges the assistance of Mr. C. D. Smith of that office in establishing the guidelines and reviewing the content of this effort.

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A POSTERIORI

The term "a posteriori" means post-experiment or "after the fact," and usually denotes knowledge (or a probability) related to or derived by reasoning from observed facts, i.e., empirical knowledge or knowledge of an event obtained from experience with that event after the occurrence of the event. For example, a decision that a target is present, based on observing the output of a sonar receiver, would be a form of a posteriori reasoning since the decision of a target's presence or absence is based on post-observation knowledge of the sonar receiver output.

A POSTERIORI OR EMPIRICAL PROBABILITY

The term "a posteriori" refers to that which is derived by reasoning from observed facts. "A posteriori probability" is therefore the PROBABILITY of an event when some or all of the outcome of the event is know or after observations of the event have been made. For example if in a number of trials an event has occurred n times and failed m times, the a posteriori or empirical probability of its occurring in the next trial is n/(n+m). It is assumed, in determining the a posteriori probability, that there is no known information relative to the probability of the occurrence of the event other than the past trials.

A PRIORI

The term "a priori" refers to that which is derived by pure reasoning and that which is (supposed to be) quite independent of experience or observed facts. In practice, it usually denotes knowledge related to or derived by reasoning without the benefit of observation, i.e., knowledge of an event obtained prior to the occurrence of that event and obtained by reasoning from definitions and assumed axioms or principles. These definitions, axioms, etc. may, however, incorporate empirical knowledge or experience of the observations of similar past events.

A PRIORI or MATHEMATICAL PROBABILITY

The term "a priori" generally means pre-experiment or "before the fact." The "a priori probability" of an event is therefore the probability of the event prior to the occurrence of the event, or before any observations of the event are made. For example, let n be the number of exhaustive, mutually exclusive, and equally likely cases of an event under a given set of conditions. If m of these cases are known as the event A, then the a priori or mathematical probability of event A under the given set of conditions is m/n. In practice, in deriving the a priori probability of an event, heavy reliance is placed on past observations of similar events. For example, the a priori probability of the presence of a submarine in a given volume of ocean at some future or present time might well be derived on the basis of the recorded past incidence of submarines in that volume of ocean. In contrast, the A POSTERIORI PROBABILITY of the submarine's presence could not be determined until some observation, say via sonar sensors, of the given ocean volume had been made, and the sonar sensors outputs examined.

See also: BAYES' RULE, A POSTERIORI PROBABILITY

ADAPTIVE FUNCTION

The concept of an adaptive function (or adaptive system) is related to the optimum response of a system to a given set of conditions. If the factors which characterize the system inputs (or those parts of the system which are not controllable) are known A PRIORI, and remain fixed, the part of the system which is controllable can be designed to meet an optimum output response criterion. If the uncontrolled parameters vary from those values upon which the optimum design was based, the system output will no longer be optimum. If it is possible to follow the temporal or spatial variation of these uncontrolled parameters, and adjust the controllable part of the system according to the optimum criterion, the system is said to be adaptive, and the function related to the controlled part of the system is called an adaptive function. This function would contain parameters which could be both time and space dependent.

ADAPTIVE SYSTEM

An adaptive system is a system having the capability to adjust to varying operating conditions, in order to maintain optimum system performance.

In sonar applications, varying environmental conditions of the ocean may cause changes in background NOISE, or PROPAGATION LOSS. In a sonar system with an adaptive capability, parameters such as gain, phase shift or time delay, and INTEGRATION TIME are automatically adjusted to give optimum values of system performance parameters, such as beamformer output SIGNAL-TO-NOISE RATIO. This implies, of necessity, that the sonar system has the capability to determine the current status of its operational environment and to act on this information.

AMBIGUITY FUNCTION

The ambiguity function is the envelope of the response of a MATCHED FILTER when the input is mismatched in DOPPLER. If $u(t)$ is the modulation envelope of the transmitted signal, the ambiguity function for time shifts τ and Doppler shifts ϕ is given mathematically by

$$X(\tau,\phi) = \int u(t)\, u^*(t+\tau)\, e^{-i2\pi\phi t}\, dt ,$$

where * denotes the complex conjugate of $u(t)$. It can be seen that $X(\tau,\phi)$ reduces to the autocorrelation of the transmitted signal $u(t)$ when $\phi = 0$. The value at (0,0) is equal to E where E is the signal energy.

The significance of $X(\tau,\phi)$ in sonar theory is that a target with range and velocity corresponding to (τ_o,ϕ_o) cannot possibly be resolved from a target at $(\tau_o+\tau,\phi_o+\phi)$ if $X(\tau,\phi)$ is equal to $X(\tau_o,\phi_o)$. However, if $X(\tau,\phi)$ is approximately equal to $X(\tau_o,\phi_o)$, resolution will be difficult.

ANGULAR RESOLUTION

The angular resolution of a sonar system is the minimum angular separation that two targets can have and yet be distinguished as separate targets.

The angular resolution of a system depends upon many factors such as the sonar array characteristics, the received SIGNAL-TO-NOISE RATIO, and the characteristics of the two targets. A nominal angular resolution for the system may be defined, however, as the angle between the maximum response axis and the first null of the receiving beam pattern (assumed to be symmetric about the maximum response axis). Thus, if two targets are separated by this nominal angular resolution, when a signal from one target is received on the main response axis, the signal from the other target will be nulled, and the two targets may usually be resolved. However, two targets producing signals of greatly different structure may sometimes be resolved even when their angular separation is less than the nominal angular resolution.

BAYES' RULE

In accepting or rejecting various hypotheses about the state of a sonar receiver output (target present?, absent?, possibly present?, etc.) the PROBABILITIES of these hypotheses, conditional upon the occurrence of the observation (or given that the observation has occurred), are of interest. These are the so-called A POSTERIORI PROBABILITIES: the probabilities of the truth of each hypothesis after the event has occurred.

Also of interest are the probabilities of these hypotheses being true prior to the observation of the event. These are the so-called A PRIORI PROBABILITIES. The conditional and the A PRIORI PROBABILITIES of the various hypotheses are related to the A PRIORI PROBABILITIES in the following manner.

Denote

$P(h_i)$ = a priori probability of the hypothesis h_i,

$P(h_i/e_k)$ = a posteriori probability of the hypothesis h_i being true given the occurrence of the event e_k,

$P(h_i,e_k)$ = PROBABILITY of the joint occurrence of h_i and e_k.

Then, by definition

$$P(h_i,e_k) = P(e_k)P(h_i/e_k).$$

Also by definition,

$$P(h_i,e_k) = P(h_i)P(e_k/h_i).$$

And third, since the event e_k can occur only if some hypothesis holds

$$P(e_k) = \sum_i P(h_i)P(e_k/h_i).$$

Combining these equations,

$$P(h_i/e_k) = \frac{P(h_i)P(e_k/h_i)}{P(e_k)} = \frac{P(h_i)P(e_k/h_i)}{\sum_i P(h_i)P(e_k/h_i)} .$$

This last equation is called Bayes' Rule, and is important in both DETECTION theory and probability theory.

BEAM PATTERN

Beam pattern is a term that describes the directional response of a transducer array. The term is used to describe the pattern of sound projector transmission as well as hydrophone reception of incident signals. The directional response of an array depends on the relative locations of the array elements, frequency of the signals being transmitted or received, and direction of arrival of signals incident on a listening array.

Beam patterns often are shown graphically as plots of SOUND PRESSURE LEVEL as a function of space about an array surface or center, for various frequencies. Beam patterns are three-dimensional phenomena, but often are shown for two-dimensional planes of interest, e.g., horizontal, vertical, etc.

BISTATIC SONAR

Bistatic Sonar is a term applied to an echo ranging system in which the sound source is located an appreciable distance from the receiving array, in contrast to the "monostatic sonar" in which the source and receiving arrays are coincident.

Bistatic sonar operations have several characteristics that differ from monostatic operation. First, in general the acoustic path length from source-to-target will not be the same as that from target-to-receiver. Thus, the total transmission loss for an echo is not simply twice the one-way loss from the source (or receiver) to the target. Second, TARGET STRENGTH values and REVERBERATION scattering strengths are usually measured or calculated in terms of sound energy that is scattered by an object and reradiated back along the angle of incidence.

Since the scattering angle usually will not correspond to the incident angle for a bistatic sonar, appropriate corrections must be made for bistatic target and reverberation scattering strength values. Finally, when either the source or the receiver of a bistatic sonar are mobile, complex system compensations must be made to account for own-ship DOPPLER FREQUENCEY SHIFT nullification and to obtain accurate target range information.

Bistatic operation offers potential advantages when the receiver can be placed in a relatively quiet NOISE environment (e.g., on a buoy) instead of the noisy background experienced by the hull mounted sonars of high speed destroyers.

BOTTOM BOUNCE

Bottom bounce refers to sound energy reflected off the ocean bottom. Sound pressure waves incident on the bottom are reflected because of the acoustic impedence mismatch between water and the bottom material. Bottom bounce propagation is an important technique used to extend acoustic detection ranges over those attanable with direct path propagation through the surface duct.

Bottom bounce transmission loss is determined by spherical spreading and attenuation (which are dependent on the path length) plus the bottom reflection loss. This bottom loss is a function of the bottom type (mud, sand, etc.), angle of incidence and angle of reflection of the sound pressure wavefront with the bottom, and the frequency of the acoustic signal. Bottom bounce transmission loss tends to be independent of range in deep water, since the increase in spreading and attenuation with range is partially offset by a reduction in bottom loss due to smaller angles of incidence and reflection. Bottom roughness and slope, and stratification of the bottom material all contribute to the success with which the bottom bounce path can be used by a sonar.

CAUSTIC EFFECT

The caustic effect in underwater sound propagation is the focusing of sound rays due to reflection or refraction under certain special conditions. The region in which the focusing occurs is called the caustic. Because of this focusing, which brings together within a small volume all the rays emitted by the source within some solid angle, the caustic effect is always associated with an abnormal increase in the intensity of sound. A CONVERGENCE ZONE is formed when a slight caustic occurs at or near the surface.

Wave theory must be used for the detailed analysis of the caustic effect, because RAY THEORY predicts that the cross-section of a bundle of sound rays becomes zero along a caustic, thus its application leads to inifinite energy density.

CAVITATION

Cavitation is the formation of bubbles or cavities surrounding microscopic gas nuclei in a fluid when the local pressure in the fluid is reduced below the pressure in the cavity. Local pressure reduction can result from flow separation around objects, the presence of vortices such as around the trailing edges of propellers, and high intensity acoustic transmission.

Cavitation bubbles continue to grow until the pressure differential between the cavity and its surrounding fluid equals the bubble surface tension. In the case of flow-induced cavitation, the bubbles are transported to regions of higher local pressures; whereupon they collapse, resulting in impulsive noise. A caustically induced cavitation ceases when the sound intensity is reduced below a critical value. Generally the onset of cavitation can be delayed by any means which increases ambient pressure.

The practical consequences of cavitation in sonar applications are several. Flow-induced cavitation is a principal source of noise for any vessel operating at high speeds. Cavitation also limits the maximum intensity at which a sonar can transmit and hence the expected source level. Further, collapsing bubbles can cause erosion of transducer faces as well as other physical structures located in areas of cavitation.

CLASSIFICATION

Classification in ASW sonar operation is the process leading to a decision as to the nature of a sonar contact. In particular it is the process for determining whether a contact is submarine or non-submarine. Classification is based on the presence or absence of a number of clues or characteristics of the signal from the contact and the weights assigned to the characteristics. Classification may occur for either active or passive sonar operation

Active sonar classification is based on the received echo and depends on the modifications that the signal undergoes in being reflected. Echo characteristics that are considered include echo strength, echo length, and echo structure, DOPPLER SHIFT, PPI pip orientation, BTR leading edge alignment, and wake echo. Not all of these characteristics may be available for a particular contact.

Passive sonar classification is based on acoustic energy radiated by the target and received by the passive sonar. Characteristics of the received signal on which classification may be based include amplitude variations. spectral line structure and envelope transients in the acoustic signature.

COHERENCE

Coherence is that property of two functions which pertains to their similarity or likeness. When two signals (functions) exhibit perfect or complete coherence they are said to be coherent, i.e., exactly alike. Obviously there are degrees of coherence and thus there is the need for measures of coherence.

The degree of similarity, as viewed in the time domain. between two functions x(t) and y(t) is generally measured by their normalized cross-correlation function, $\zeta_{xy}(\tau)$. This function is defined by

$$\zeta_{xy}(\tau) = \frac{E[x(t + \tau)y(t)]}{\sqrt{E[x^2(t)]\ E[y^2(t)]}}$$

where stationarity in the second order statistics is taken to hold. In more familiar notation, the expectation E[ ] operations can be denoted by unnormalized correlation functions. Thus

$$E[x(t + \tau)y(t)] = R_{xy}(\tau)$$

$$E[x^2(t)] = R_{xx}(0)$$

$$E[y^2(t)] = R_{xy}(0)$$

Now $\zeta_{xy}(\tau)$ takes on a more conventional form,

$$\zeta_{xy}(\tau) = \frac{R_{xy}(\tau)}{\sqrt{R_{xx}(0) Ryy(0)}}$$

The physical interpretation of $\zeta_{xy}(\tau)$ is that it measures the likeness of x(t) and y(t) at differenct values of relative delay between x(t) and y(t) and does this in a manner which is independent of the powers in either x(t) or y(t).

Another way, albeit much less commonly used, of measuring the degree of coherence between x(t) and y(t) is the coherence function $\gamma^2_{xy}(f)$. This measure of similarity is defined in the frequency domain as

$$\gamma^2_{xy}(f) = \frac{|G_{xy}(f)|^2}{G_{xx}(f)\ G_{yy}(f)}$$

where the cross-power spectrum is defined by

$$G_{xy}(f) = \int_{\infty}^{\infty} R_{xy}(t)\,e^{-j2\pi ft}dt$$

and similarly for the auto-power spectra $G_{xx}(f)$ and $G_{yy}(f)$. The physical interpretation of $\gamma^2_{xy}(f)$ is that it reveals at each frequency, f, the fractional likeness (on a scale from zero to one) of x(t) and y(t), and does so in a way that is independent of the power of x(t) and y(t) at the frequency in question.

The normalized cross-correlation function is sensitive to the relationship between x(t) and y(t) as well as being sensitive to the coherence (similarity) in the waveform structure of x(t) and y(t). The coherence function $\gamma^2(f)$, on the other hand, is not sensitive to the phase relationship between components of x(t) and y(t).

CONSTANT FALSE ALARM RATE

Constant false alarm rate refers to a uniform number of FALSE ALARMS per unit time.

The rate at which the FALSE ALARMS occur is dependent on the THRESHOLD level of the receiver: the higher the THRESHOLD, the fewer the number of FALSE ALARMS. In choosing a THRESHOLD level, consideration must be given to the problem of raising the level so high that the presence of some signals cannot be detected. In order to overcome this difficulty, threshold adjustment to maintain a constant false alarm rate can be made by an operator or can be controlled automatically. One form of automatic threshold adjustment in common usage utilizes the average background noise level for control.

CONSTANT FALSE ALARM RECEIVER

A constant false alarm receiver (CFAR) is a receiver in which the detector THRESHOLD is adjusted in such a way as to keep the false-alarm rate constant. (In a receiver having a fixed preset detection threshold, the presence of excessive noise, due to increases in environmental noise or countermeasures jamming, could increase the FALSE ALARM RATE to an intolerable extent.) A CFAR solves this problem at the expense of decreasing the PROBABILITY of DETECTION in the presence of higher noise levels. Threshold adjustment in a CFAR may be either manual or automatic.

CONVERGENCE ZONE

Convergence zone is the term applied to a circular annulus in the ocean about an acoustic source where vertical refraction produces a strong focusing (or convergence) of the emitted sound pressure wave. This focusing of energy creates a relatively intense SIGNAL FIELD in the convergence zone annulus compared to the field that would be produced by simple spherical spreading of sound.

The difference between a sound field intensity LEVEL due to spherical spreading and that due to convergence zone focusing is often called the convergence gain, which typically averages 30 dB for shallow source and receivers at long range. Refraction effects actually cause the convergence zones to repeat at successive range intervals to form a set of concentric annuli about the source, thereby giving rise to the terms 1st convergence zone, 2nd convergence zone, etc. Typically, the width of the zones extends over a range of 1-3 miles.

The range to the zones, zone width, and the convergence gain are determined entirely by the depth of the acoustic source and the sound velocity profile. Thus, these characteristics can be expected to show a seasonal and a geographical area dependence It should be stressed that the convergence zone is produced solely by the refraction effects and not by depressing the axis of a directional sonar array. Depressing the sonar beam only determines the intensity of the sound that is directed into the convergence channel.

CORRELATION FUNCTION

Correlation is a comparison process carried out between two signals to determine their similarity to each other or to show a causal relationship between them. The correlation between two functions of time, $f_1(t)$ and $f_2(t)$, is a combination of three operations, displacement, multiplication, and averaging (or integration), that yields a measure of their statistical dependence. The result of correlation is, in general, a function of the amount of the displacement in the first operation, and of the kind of averaging process used in the third operation, and is called the correlation function. When the two functions are the same, one speaks of the autocorrelation function, and when they are different and one wants to stress that fact, one speaks of the cross-correlation function.

The correlation functions most frequently used in sonar applications are the functions obtained when $f_1(t)$ and $f_2(t)$ describe random processes taken from stationary ensembles, and when the averaging is done over all of time. In that case, the correlation function is given by

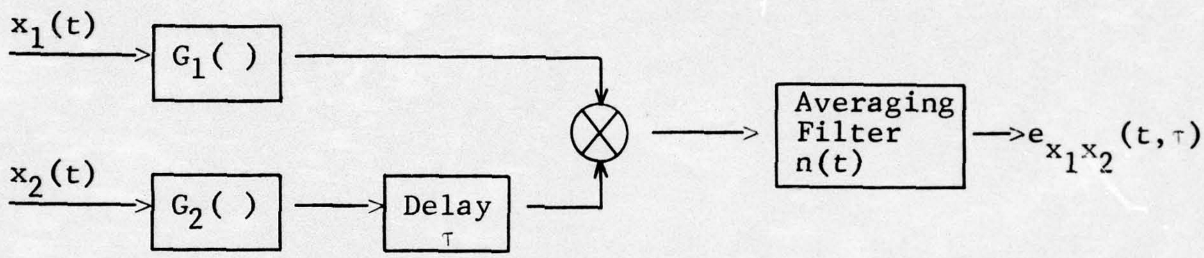
$$C(\tau) = \lim_{T \to \infty} (1/T) \int_{-\frac{1}{2}T}^{\frac{1}{2}T} f_1(t)\, f_2(t+\tau)\, dt$$

where τ is the amount of displacement in time, and where it is assumed that this limit exists.

If $f_1(t)$ and $f_2(t)$ are the same function, the result is the autocorrelation function; if $f_1(t)$ and $f_2(t)$ are different the result is the cross-correlation function. In either case, if the correlation function is plotted, as a function of τ, the result is called a correlogram. Correlation resolution, then, is the ability to exhibit fine detail in the correlogram, the purpose being the ability to distinguish between two closely separated points (targets) in space. This ability to portray fine detail is a function of how small the value of the steps in τ may be taken.

CORRELATORS

A correlator is a signal processing device that, in operation, produces estimates of the correlation functions of pairs of input waveforms, $x_1(t)$ and $x_2(t)$. There are two broad types of correlators, known as cross-correlators and replica correlators, differing primarily in the means by which $x_1(t)$ is obtained. Either type of correlator may be represented by the following block diagram:



The cross-correlator, most often encountered in passive sonar applications, forms an estimate of the cross-correlation function between the input time functions according to the following, quite general, functional form:

$$C_{x_1x_2}(t,\tau) = \int_{-\infty}^{t} h(t - s)G_1\big(x_1(s)\big)G_2\big(x_2(s + \tau)\big)ds$$

where

$h(t)$ = impulse response of the integrating filter--the final stage of any correlator, e.g., an RC filter in which case

$$h(t) = \frac{1}{RC} e^{-t/RC} .$$

$G(x)$ = any zero memory amplitude mapping on the function $x(t)$. If we are dealing with clipped correlators, the $G(x) = \text{sgn}[x]$, while for linear correlators $G(x) = x$, and

τ = a time shift between $x_1(t)$ and $x_2(t)$.

In practice, cross-correlators produce only an estimate of the correlation function because, strictly speaking, the averaging process, $\int_{-\infty}^{t} h(t - s)\cdot(\ )ds$, occurs only over a finite duration, rather than over the arbitrarily long interval included in the definition of the correlation function.

Correlators as shown in the block diagram have been built for several sonar systems. In one application input waveforms $x_1(t)$ and $x_2(t)$ are obtained from so-called split beam outputs which are scanned simultaneously in azimuth, thus forming estimates of the correlation function for waveforms arriving from different points in space. In this case the delay τ is actually introduced by the medium, geometry and array. Another near operational passive correlation system obtains the $x_1(t)$ and $x_2(t)$ signals from preformed beams and then clips $\left(G_2(\ ) = \text{sgn}(\ )\right)$ and forms several hundred cross-correlation coefficients (values of the correlation function) for several hundred values of τ.

The second general type of correlator is the replica correlator which is found in active sonar applications. In this case, the form of $C_{x_1x_2}(t,\tau)$ becomes slightly different because, say, $x_1(t)$ is a stored replica of the transmitted waveforem. That is $x_1(t) = r(t)$. As before the input waveforms may be clipped or linear. Thus when the reference is clipped we have $G_1\left(x_1(t)\right) = \text{sgn}\left(r(t)\right)$. The mathematical expression for the replica correlator is

$$C(t) = \int_{t-T}^{t} x(t - s)\,\text{sgn}[r(s)]\,ds$$

The correlation, in the active case, is the time domain equivalent to the matched filter which is the optimum receiver for detecting known signals in a background of white GAUSSIAN NOISE.

In the passive case, when $x_1(t)$ and $x_2(t)$ both contain a common element, namely the signal, and when the relative delay is such that the signal in both channels is time aligned the output becomes proportional to the SQUARE OF THE SIGNAL plus the cross-correlation of the noise which is small if the noise components in $x_1(t)$ and $x_2(t)$ are independent.

Hence, in both cases correlation produces signal enhancement.

The term auto-correlation function is used to discuss the correlation function obtained when $x_1(t) \equiv x_2(t)$.

DECIBEL

The decibel is a measure of LEVEL, expressed as a ratio of intensities, and stated in logarithm to the base 10. If P and P_o are two power quantities, the level (L) of P relative to P_o is expressed in decibels (dB) as

$$L = 10 \text{ Log } (P/P_o).$$

Use of the decibel measure provides a convenient method for expressing large changes in parameters; it further simplifies the arithmetic in multiplying quantities stated in dB, by merely adding their decibel values.

Examples of power quantities used in sonar technology that qualify for expression in decibels are SOUND PRESSURE squared, particle velocity squared, sound intensity, sound energy density, and voltage squared.

See also: LEVEL

DECISION CRITERIA

Decision criteria are the rules by which men and machines make decisions regarding the acceptance of statistical hypotheses. For sonar, these hypotheses may be whether a target is present, whether the target is a submarine, and whether the submarine is an enemy. These rules are applied to the observations or data supplied to the man (operator) or machine by the sonar, and are such that if the data behave in a prescribed fashion then one or more of the hypotheses is accepted.

Clearly the complexity of the decision criteria becomes greater as the operator seeks to make more sophisticated decisions. Thus, the necessity for aiding the operator in making his decision increases as the problem becomes more complex -- i.e. complex from a decision making point of view. These considerations have led to the development of some automatic DETECTION machines, little development in automatic CLASSIFICATION and even less effort in the area of automatic identification.

DEEP SCATTERING LAYER

The deep scattering layer is a horizontal concentration of acoustic energy scatterers. These scatterers are bathypelagic fish having internal air bladders of varying sizes. Deep scattering layers have an adverse impact on sonar performance because of their contribution to volume REVERBERATION.

Deep scattering layers may range from 150 to 600 feet in thickness. Many layers migrate vertically with a diurnal cycle, with some components of the layer rising to a depth of less than 400 feet at night and descending to a depth of from 600 to 2000 feet during daylight. The rate of ascent of a layer may be as high as 600 feet per hour.

Scattering strength within a layer is almost constant and is dependent on layer depth. Scattering strengths of deep scattering layers show large seasonal variations, with highs in March and April and lows in July through October. Scattering strength also varies with latitude.

DELTA FUNCTION

The delta function, $\delta(x)$, or Dirac delta function is a function used to express an impulse of unit area occuring at a discrete point in time, frequency or space. Formal properties assigned to the delta function are infinite height (magnitude) and infinitesimally small width, (duration).

$$\delta(x-x_o) = 0 \qquad (x \neq x_o)$$

$$\int_{-\infty}^{\infty} \delta(x-x_o)dx = \int_{x_o-\epsilon}^{x_o+\epsilon} \delta(x-x_o)\, dx = 1 \quad \text{for all } \epsilon > 0$$

$$\int_{-\infty}^{\infty} \delta(x-x_o)\, F(x)\, dx = F(x_o)$$

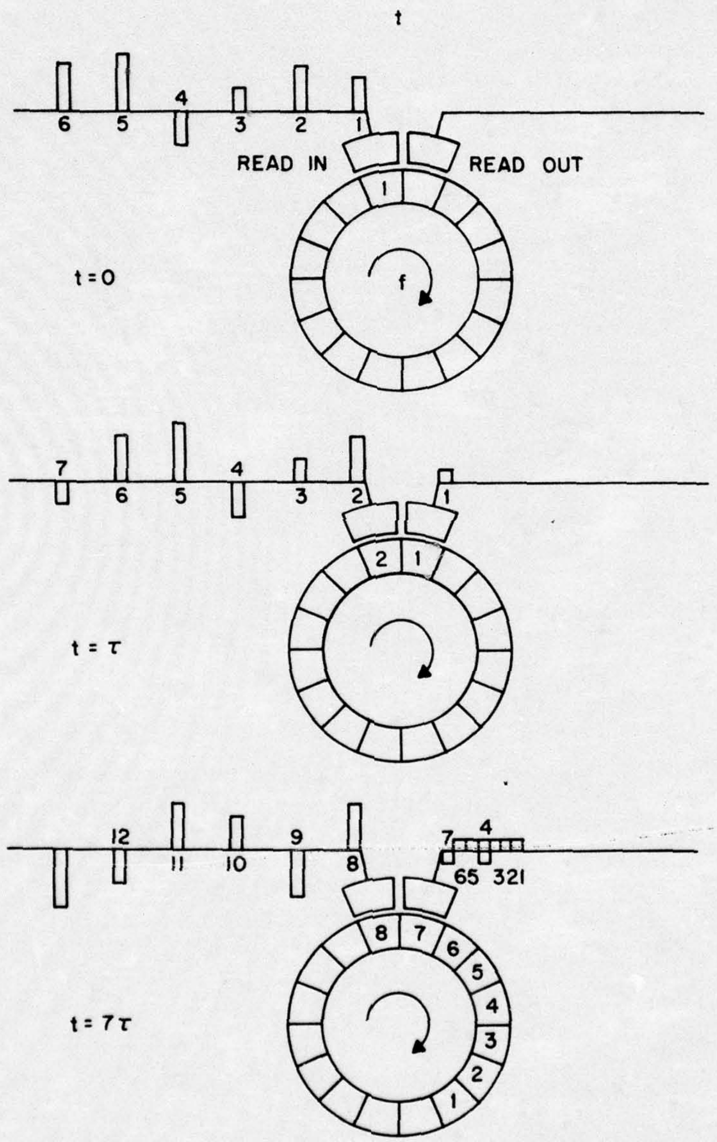
DELTIC

A DELTIC is a device for achieving TIME COMPRESSION. It is an acronym for DElay Line TIme Compressor. The implementation of a DELTIC can be achieved by means of a magnetic drum, a recirculating delay line or by means of a digital shift register. The storage medium stores sequential samples from a section of waveform with length T. The storage medium advances the positions of the sample at a rapid rate. They emerge from the medium and are reinserted at the other end of the medium. The samples are updated each τ seconds (the time between waveform samples), the oldest sample in the storage being replaced by a new sample during every τ recirculation. During the recirculation the storage medium "reads out" all the samples in the T seconds of waveform every τ seconds. Thus T seconds of waveform may be examined by the processor every τ seconds. The time compression ratio is T/τ.

Although time compression is most often realized in practice by the use of a recirculating delay line or a shift register, the mechanism is perhaps most easily described by means of a magnetic drum implementaion (see the included diagram).

The operation is shown in three stages. At time $t = 0$, the first sample is entered on the drum. At time $t = \tau$, the drum has completed slightly more than one revolution (one revolution is completed at time $t = \tau - \epsilon$) and the second sampled pulse is being entered while the first is being read out. At time $t = 7\tau$, seven old samples have been replaced. When the drum is filled with N samples (which is really the condition of primary interest), the oldest sample is discarded and a new sample is added at each revolution. The N samples are sequential real-time samples of

duration $T = N\tau$. Thus, it is evident that during slightly more than one revolution of the drum, or τ seconds, information gathered over a period equal to $T = N\tau$ seconds is available for processing.



See also: DELTIC CORRELATOR

DELTIC CORRELATOR

The DELTIC correlator is one example of the use of a DELTIC (DElay Line TIme Compressor) in a processor. The DELTIC correlator contains two delay lines; the signal delay line and the reference delay line. The time between samples of the real time waveform is τ.

The signal delay line is a recirculating delay line containing T seconds worth of input signal plus noise waveform. There are $N = T/\tau$ samples in the delay line. In τ seconds all N samples are read out of the delay line and the oldest sample is replaced by a new sample.

The reference delay line also contains N samples of the reference function which is a scaled replica of the transmitted signal. These samples are read out in time τ, but no change in the contents is made.

Each τ seconds the contents of the two delay lines are multiplied sample by sample and the N products are summed to form a single sample at the output of the correlator. The next sample at the output of the correlator is formed in the same way except that all of the signal plus noise samples are shifted one sample time τ relative to the reference function.

See also: DELTIC, HETERODYNE CORRELATOR, MATCHED FILTER, and CORRELATOR

DETECTION

Detection is the process of determining the presence of a signal in a noise background. In sonar technology, the sonar receiver is used to process the acoustic observations to aid the human observer in making his decision of whether or not a target is present. Since detection is a decision process, then the PROBABILITY of making a correct decision becomes a useful measure in predicting the detection performance of a human observer/sonar receiver. Also, the probability of calling signal-present in a noise only background (FALSE ALARM probability) and the probability of calling signal-absent in a signal-present situation (missed probability) are used in predicting detection performance.

Note: It is unfortunate that the term detection is also used to refer to the demodulation process in communication systems (i.e. extracting the amplitude modulation signal from a received waveform in an amplitude-modulation (AM) receiver). Demodulation is a more descriptive term for this type of processing, even though the terms detection and demodulation are carelessly used interchangeably.

DETECTOR

A detector is a circuit element which is used to extract modulation information from an input signal. The forms of the modulation used in communication equipment include amplitude modulation, frequency modulation, phase modulation, and pulse code modulation. In sonar applications amplitude modulation is of primary interest. Several forms of non-linear detectors are used for extracting amplitude modulation. In its simplest form the non-linear device provides an output function r(t) in response to a stimulus s(t) which may be represented by

$$r(t) = |s^n(t)|$$

Such a device is called an _n_th law detector. The values n = 1 and n = 2 are often used. When n = 1, the detector is referred to as a linear detector. When n = 2, the detector is referred to as a square law or a power detector. The detector followed by its averager is referred to by one of two names, detector or detector-averager, depending upon the value of n and the averaging time.

The term coherent detector is also in common use. It is a complete signal processor which includes MATCHED FILTERING prior to envelope detection. Coherent detectors are MATCHED FILTERS such as the synchronous detector and the CORRELATOR. In the HETERODYNE CORRELATOR implementation of a coherent detector, the predetection circuitry maximizes the SIGNAL-TO-NOISE RATIO in the waveform which is submitted to a _n_th law detector. There is no further PROCESSING GAIN available in the mildly averaged detector output.

DIRECTIVITY

The directivity of a transducer or transducer array, used either as a transmitter or as a receiver, is its property of concentrating the radiated energy in one or more particular directions (for a transmitter), or of being more sensitive to received energy from one or more particular directions for a receiver.

The directivity of a sonar device, consisting of one ore more transducer elements, is limited by the ratio of a characteristic length of the device to the dominant wavelength in the energy spectrum of the signal. A device whose characteristic length is small relative to the wavelength can have very little directivity; consequently, a directive device has a characteristic length which is large relative to the wavelength. A line hydrophone, for example, can be highly directive in the plane containing its axis, and virtually non-directional in a plane perpendicular to its axis. For a device of given dimensions, the directivity achieved depends on the SPATIAL PROCESSING, i.e., beam forming, procedure used in transmitting or receiving acoustic energy.

A quantitiative measure of directivity is specified by the farfield BEAM PATTERN*. For a transmitter, the farfield beam pattern specifies the angular distribution of acoustic intensity, i.e., the radial energy flux, crossing a sphere of large radius enclosing the device. For a receiver, the BEAM PATTERN specifies the output voltage for a plane wave incident

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*Synonomous terms include "radiation pattern," farfield directivity" pattern, and "directivity" pattern

on the device from an arbitrary angle. It is a consequence of the reciprocity principle that the transmit and receive BEAM-PATTERNS of a device used both as a transmitter and as a receiver are identical if the SPATIAL PROCESSING methods for transmitting and receiving are reciprocal.

For many applications, identical transmit and receive beam patterns are not desired. In active, scanning sonars, for example, an ommidirectional transmitting pattern is used, along with a directional receiving pattern.

DIRECTIVITY INDEX

The directivity index of a transducer, in decibels, is 10 times the logarithim to the base 10 of the directivity factor. The directivity factor is the ratio of the sound pressure squared at some fixed distance and in some specified direction to the mean-square sound pressure at the same distance averaged over all directions from the transducer. The distance must be great enough so that the sound appears to diverge spherically from the effective acoustic center of the sources. Unless otherwise specified, the reference direction is understood to be that of maximum response.

The directivity factor of a transducer used for sound reception is the ratio of the square of the open-circuit voltage produced in response to sound waves arriving in a specified direction to the mean squared voltage that would be produced in a perfectly diffused sound field of the same frequency and mean squared sound pressure.

DOPPLER SHIFT

A Doppler Shift is the frequency shift imposed on a signal by the relative motion of the signal sensor and the target. The difference in frequency between an echo reflected from a motionless object and an echo refelected from a moving target is a manifestation of the Doppler effect. The presence of a Doppler shift in an echo (implying target motion) has traditionally been one of the most important active sonar CLASSIFICATION clues. The existence of a discernible Doppler shift in an echo usually is determined by a comparison of the target echo frequency with the center frequency of the REVERBERATION, the latter being a reflection of acoustic energy from essentially motionless ocean inhomogeneities and boundaries.

Doppler shift is proportional to both the frequency of the transmitted signal and the relative velocities of the target and sensor in the direction of the sound beam. For active sonar, the Doppler frequency is given by the equation

$$f_D = f_o \left[\frac{C - (v_t - v_s)}{C + (v_t - v_s)} \right] \simeq f_o \left[1 - 2\Delta v / c \right]$$

where f_D = received, Doppler-shifted echo center frequency

f_o = transmitted center frequency

v_s = component of own ships' velocity along the line joining sensor and target

v_t = component of target's velocity along the line joining sensor and target

c = speed of sound in the water.

Δv = rate of change of separation between the transducer and target, i.e., the range rate.

EXPECTATION OF A FUNCTION OF A RANDOM VARIABLE

The expectation or average of a function of a random variable is a weighted summation. The weights are the probability that the RANDOM VARIABLE will take on that value. If u(x) is a function of the RANDOM VARIABLE, x, and E denotes the expectation, then

$$E(u(x)) = \int_{-\infty} u(x)\ p(x)\ dx$$

where p(x) is the PROBABILITY DENSITY function of x.

Two common expected values are the mean,

$$\mu = E(x) = \int_{-\infty}^{\infty} xp(x)dx$$

which measures the central location of a RANDOM VARIABLE, and the variance, σ^2,

$$\sigma^2 = E(x-\mu)^2 = \int_{-\infty}^{\infty} (x-\mu)^2 p(x)dx$$

$$= \int_{-\infty}^{\infty} x^2 p(x)dx - \mu^2 = E(x^2) - \mu^2$$

which measures the dispersion of the RANDOM VARIABLE around its central location.

FALSE ALARM

Given a DETECTION CRITERION, a false alarm occurs whenever the criterion is satisfied by a stimulus that was supplied when a target was not present. This can occur even in a DETECTION, CLASSIFICATION, or identification context. It should come as no surprise to find that the stimuli arising from NOISE alone can satisfy the DETECTION rule, otherwise there would be no problem whatsoever in always choosing the correct hypothesis.

The words "FALSE ALARM" and "DETECTION" have often been misused because they have appeared in an immense body of literature dealing with the performance of machine detection of various signal processors. The implication here is that of machine detection, such as simple THRESHOLD crossings for noise alone and signal plus noise. This does not include the DETECTION process which characterizes the human observer. A "false alarm" is a random event. That is, random noise can satisfy the detection criterian, and be called a signal. It follows then that false alarm behavior (as with DETECTION behavior) is most meaningfully specified or quantified by FALSE ALARM PROBABILITY*. Hence when dealing with humans, it is usually necessary to talk in terms of their FALSE ARARM PROBABILITY as derived from a controlled experiment.

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*Other measures such as false alarm rate, (FAR) are sometimes used because of their closer relationship to tactical or even strategic considerations.

FALSE CONTACT

A false contact is said to have occurred when the operator detects a target-like stimulus and subsequently classifies it as a non-submarine. False contacts may arise from whales, kelp, turbulent regions, and submerged pinnacles or wrecks. In some usages, surface ships may be considered false contacts.

FINE STRUCTURE

The fine structure of a signal is the detailed characteristics of its amplitude, frequency and phase structure that may give information about the size, shape, differential DOPPLER SHIFT and reflecting characteristics of the target.

Echo fine structure in general will be a function of the aspect angle, MULTIPATH structure, size, shape, speed, turning rate, presence of hull and wake echoes, and reflecting characteristics of the target. The fine structure may be examined only if the range, angular, and Doppler resolution of the sonar are sufficient to allow separation of the reflections from different parts of the target or recognition of reflectors with different DOPPLER SHIFTS.

Fine structure, when present, is the primary tool used in target CLASSIFICATION, but it also greatly aids the DETECTION process.

FIRST ORDER AND HIGHER ORDER EFFECTS

In expressing a dependent sonar parameter in terms of an independent sonar parameter, the following general expression can usually be written as

$$
\begin{aligned}
y &= f(x) \\
&= 1 + ax + bx^2 + \ldots
\end{aligned}
$$

where

y = dependent sonar parameter,
x = independent sonar parameter.

The x term is referred to as a first order effect. The higher powers of x are referred to as higher order effects. If the approximation $x \ll 1$ is made, then the first order term, x, is the only pertinent term in the expression for y.

GAUSSIAN DENSITY FUNCTION

The Gaussian, or Normal probability density function is important because it adequately describes distributions which appear frequently in nature. Mathematically, the Gaussian density function, g(x), is given by

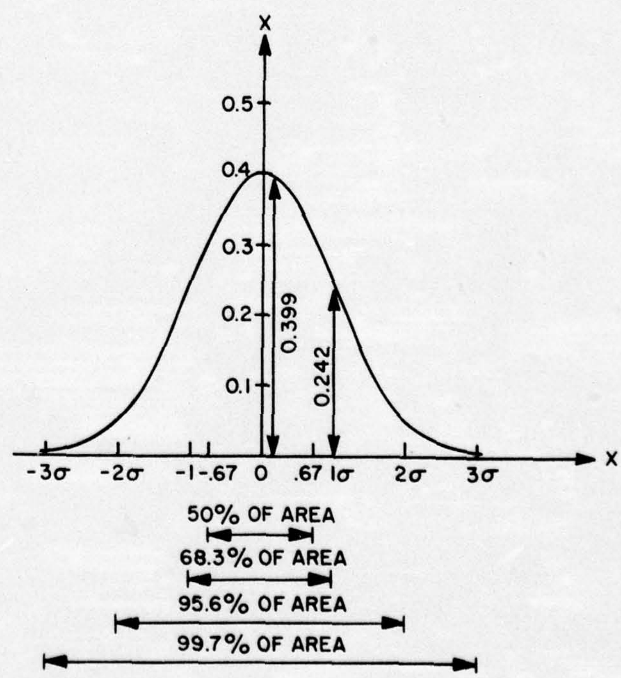
$$g(x) = \frac{1}{\sqrt{2\Pi}\,\sigma} \exp\left\{-(x-\mu)^2/2\sigma^2\right\}$$

where

$$\mu = E(x)$$

$$\sigma^2 = E\,(x-\mu)^2$$

The graph of this function is the familiar "bell-shaped curve," as shown here. Most statistics books provide tables based on this function.



When data fall into a Gaussian distribution, the standard deviation, σ, is an indicator of how tightly the data points cluster about the mean value. With a perfect Gaussian distribution, about 68% of the data points will lie within $\pm 1\sigma$ of the mean value, over 95% within $\pm 2\sigma$, and 99.7% within $\pm 3\sigma$.

HETERODYNE CORRELATOR

A heterodyne correlator is a form of DELTIC correlator. Heterodyne correlators can be implemented in three ways, described as the sum-band, the difference-band, and the D.C. heterodyne (or quadrature) correlators.

In the difference-band correlator described here, the reference signal is a replica of the transmitted pulse. For a single propagation path, the echo from a point scatter on the target is an amplitude-scaled, delayed, Doppler-shifted version of the transmitted pulse. If the transmitted signal has the form

$$A_o \sin 2\pi ft,$$

the return signal will have the form

$$kA_o \sin 2\pi f(1+r)(t-\delta),$$

where f is the transmitted frequency, $r \ll 1$ is the fractional Doppler shift, and δ is the time of travel to and from the target.

When the reference signal waveform is delayed to match the travel time δ, the product of the echo and signal waveforms is

$$kA_o^2 \left[\cos 2\pi fr(t-\delta) - \cos 2\pi f(2+r)(t-\delta)\right]$$

The frequency, fr, in the first cosine term is the observed Doppler shift, i.e., the difference in frequency between the echo and the transmitted frequency. The frequency f(2+r) is the sum of the frequencies in the transmitted and the echo signals. Both these frequency components appear at the output of the multiplier in the DELTIC correlator.

In the difference-band correlator, a filter bank with channel widths matched to the Doppler resolution is used to analyze the frequency components which could be expected in the difference frequencies; the sum band components are rejected by the filter bank. For this example, the frequency variation was made simple (frequency constant). It is easily seen, however, that even if f depends upon time, the difference frequency in the product samples is independent of time as long as the reference signal is identical to the echo except for Doppler shift.

The sum-band correlator with a constant frequency signal could be implemented by placing a comb filter bank centered near twice the carrier frequency, that is, in the sum frequency region f(2+r). If the signal has variable frequency as in the FM sweep, however, frequency would not be constant. The sum frequency is made constant by spectrum inverting the signal in the reference delay line, so that the reference signal frequency decreases when the transmitted frequency increases; the sum of the two frequencies then remains constant. This technique provides a constant frequency in the sum band for all types of signals.

In these correlators the waveforms to be processed are usually heterodyned to a low-frequency processing band for economy of sampling. In order to preserve phase as the waveforms pass through zero frequency, quadrature of both the reference and the echo waveforms must be sampled. This means that the two waveforms must be shifted in phase by 90° and sampled in both the phase-shifted and the nonphase-shifted versions. In the correlation process products must be formed in the phase-shifted versions of the echo and reference and in the nonphase-shifted versions of the echo and reference. The comb filters in the heterodyne correlators provide the summing step required in the computation of correlation functions.

See also: DELTIC, DELTIC CORRELATOR, MATCHED FILTER, and CORRELATOR

HOMOGENEOUS MEDIUM

A homogeneous ocean medium for sonar purposes implies that the physical properties of the ocean affecting the transmission of sound are constant throughout the volume being considered. In particular, homogenous implies that the temperature, density and composition of the ocean are everywhere the same throughout this volume. The assumption of homogeneity is frequently made for purposes of theoretical analysis, but in practice the ocean medium is always non-homogeneous to some degree.

INTEGRATION TIME

In any processing system, that time interval used to accumulate or sum a variable is referred to as integration time. Most sonar systems contain circuitry (normally just prior to the display) that accumulates or sums the incoming information voltage. This summation may be affected by the implementation of several different analog (e.g., RC averaging circuitry) and digital (e.g., the arithmetic sum of the input samples) techniques. The reason for this summing process is to reduce the standard deviation of a random voltage relative to the mean of the voltage. The summing process also reduces the amount of noise variation in this voltage without degrading the signal, thus enhancing the SIGNAL-TO-NOISE RATIO.

INVERSE FILTER

An inverse filter is a filter whose gain is inversely proportional to the spectral amplitude of a prescribed input waveform, Q(t). The filter is matched to a waveform Q(t) in such a way that all frequency components of Q(t) are brought into phase coincidence and reinforce each other at a particular time instant t = T. Mathematically, if $\Phi(f)$ is the Fourier transform of Q(t), the frequency TRANSFER FUNCTION of the inverse filter is given by

$$H(f) = \frac{e^{-j2\pi fT}}{\Phi(f)}$$

The inverse filter contrasts with the MATCHED FILTER, wherein the phase coincidence also occurs but the filter gain is directly proportional the the amplitude of Q(t). The effect of an inverse filter is to suppress the stronger components of the input signal Q(t), and thus the inverse filter would be used to discriminate against the waveform Q(t) in a noise background.

LEVEL

The level of sound wave intensity is expressed in decibels above or below a stated reference intensity. For example the level of sound wave intensity I referred to I_o would be stated as

$$L(dB) = 10 \log I/I_o.$$

There are several acoustical parameters expressed in terms of levels relative to conventional reference values. Below are listed brief definitions for some typical sonar parameter levels.

(a) BAND PRESSURE LEVEL. The band pressure level of a sound for a specified frequency band is the sound pressure level for the sound contained within the restricted band. The reference pressure must be specified.

(b) INTENSITY LEVEL. The intensity level, in decibels, of a sound is 10 times the common logarithm of the ratio of the intensity of this sound to the reference intensity. The reference intensity must be stated explicitly.

(c) NOISE LEVEL. Noise level is the level of noise, the type of which must be indicated by further modifier, or context.

(d) PEAK LEVEL. The peak level is the maximum instantaneous level that occurs during a specified time interval. Peak sound pressure level is understood, unless otherwise specified.

(e) POWER LEVEL. Power level, in decibels, is 10 times the logarithm (common) of the ratio of a given power to a reference power. The reference power must be indicated.

(f) SOUND PRESSURE LEVEL. The sound pressure level of a sound, in decibels, is 20 times the common logarithm of the ratio of the pressure of the sound to the reference pressure. The reference pressure must be explicitly stated. Unless explicitly stated otherwise, effective (rms) sound pressure is understood. Note: In many sound fields, the sound pressure ratios are not the square roots of the corresponding power ratios.

(g) SPECTRUM LEVEL (SPECTRUM DENSITY LEVEL). The spectrum level of a specified signal at a particular frequency is the level of that part of the signal contained within a band 1 cycle per second wide, centered at the particular frequency. Ordinarily this has significance only for a signal having a continuous distribution of components within the frequency range under consideration. The words "spectrum level" cannot be used alone, but must appear in combination with a prefatory modifier; e.g., pressure, velocity, voltage.

See also: DECIBEL

LIKELIHOOD RATIO

Likelihood ratio is the ratio of two conditional PROBABILITY DENSITY FUNCTIONS, one for signal and noise present, and the other for noise alone. If these conditional probability density functions are respectively denoted by $p(y|S + N)$ and $p(y|N)$, where y is the sonar system output variable upon which a detection decision is to be based, the likelihood ratio is expressed as

$$L_r(y) = \frac{p(y|S + N)}{p(y|N)}$$

The likelihood ratio is a measure of how likely it is that a given output, y, is due to signal plus noise, compared with it being due to noise alone. If the likelihood ratio is suitably large, the decision is made in favor of a signal being present. Usually a threshold value of likelihood ratio is selected *a priori*, and if the likelihood ratio exceeds this threshold a decision is made that a signal is present. The likelihood ratio can be a useful statistical indicator for the description of several DETECTION criteria.

In sonar, the likelihood ratio may be used in a detector as a basis for deciding whether or not a signal was received from a target. A THRESHOLD is selected and the likelihood ratio is computed. If the likelihood ratio exceeds the threshold, the decision is made that a target is present. Otherwise, the decision is made that no target is present. The selection of the threshold depends on the relative importance attached to either missing a target or producing a FALSE ALARM.

LOCALIZATION

Localization is the process of determining the position of a target in the ocean relative to a reference point, either on a fixed or mobile coordinate system, at a particular instant in time. Localization follows DETECTION of a signal representing a potential target, and precedes TRACKING of the target. A target can be localized within certain limits dependent on the resolution capability of the sensor system used to process localization information. A localization position is normally specified in terms of range, bearing, and depth related to the sensor position.

A target can be localized by either active or passive sensors. Active sensors routinely yield range, bearing and sometimes depth by determination of travel time, horizontal angle of signal return and signal structure due to multipath propagation. Passive sensors routinely yield target bearing by measuring horizontal angle of signal arrival. Various passive ranging techniques are available which combine this bearing data over some time interval with assumptions concerning target motion and environmental parameters to yield passive estimates of target range. These techniques can be broadly classed as manual or automated horizontal or vertical plane techniques (see RANGING). Further target range can be determined passively by correlation techniques, utilizing cross-correlation of signals from several sensors.

MATCHED FILTER

A matched filter is a linear processing network which is matched to its input signal and noise waveforms in such a way as to produce the maximum SIGNAL-TO-NOISE RATIO at its output when the input is a combination of the desired signal and additive noise. While passing signal energy, these filters reject as much noise energy as possible. Matched filters are also called North filters, optimum filters, and coherent processors. In active sonar signal processing, the matched filter is frequently used in two forms: the narrow bandpass filter for CW (single frequency) pulses, and the replica correlator for FM signal waveforms.

Mathematically, the matched filter can be described in terms of its frequency response function, H(f), which for spectrally flat GAUSSIAN NOISE is given by the following expression:

$$H(f) = G\, S^*(f)\, e^{-j\omega t_1}$$

where

$$S(f) = \int_{-\infty}^{\infty} s(t)e^{-j\omega t}dt = \text{Fourier transform of the input signal,}$$

$S^*(f)$ = complex conjugate of $S(f)$,

t_1 = time at which the signal is observed,

and G = filter gain constant.

For the same conditions, the maximum output peak signal-to-mean noise power ratio of a matched filter can be shown, independent of signal waveform, to be equal to:

$$\frac{S}{N_o} = \frac{2E}{N_o}$$

where E = signal energy, and

N_o = noise spectrum level.

In practice a matched filter is seldom achieved because insufficient knowledge of the exact signal form is available to describe the required filter. In attempting to provide matched filters for sonars, designers have introduced replica correlators, as mentioned above, such as the DELTIC CORRELATOR; these are matched filters for the signals having the form of the transmitted pulse in white Gaussian noise. The echoes observed, however, are overlaid, amplitude-scaled, differentially-Doppler-shifted, and perhaps phase-modulated versions of the transmitted signal differing in detail from one echo to the next. Clearly, then, filters matched to the transmitted signal do not satisfy the criterion for maximizing the output signal-to-noise ratio, and are therefore less than optimum in practice.

MODIFIED RECEIVER OPERATING CHARACTERISTICS (MODROC)

A receiver operating characteristic or curve (See ROC) is a plot of PROBABILITY OF DETECTION versus the PROBABILITY of FALSE ALARM (or occurrence of a noise mark). This type of curve is inadequate for processor comparison since it fails to account for the rate at which independent NOISE samples occur at the output of the processors.* For this reason, a Modified Receiver Operating Characteristic or Curve (MODROC) has been developed. The MODROC is a plot of PROBABILITY OF DETECTION versus FALSE ALARM RATE (or the rate of occurrence of noise marks).

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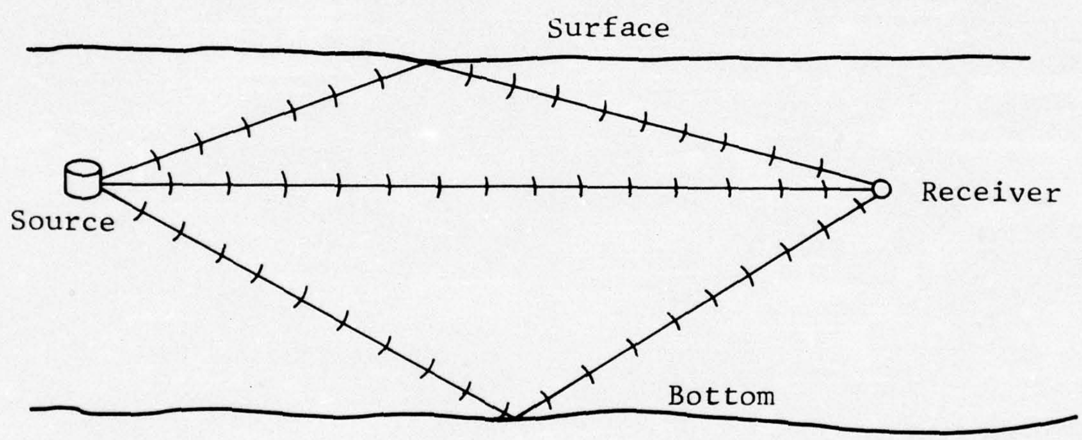
*False alarm rate, FAR, at the output of a processor is related to the probability of a false alarm, P(FA), by the relation FAR = B · P(FA), where B is the processor output BANDWIDTH and is equal to the number of independent noise samples per second that occur at the processor output.

MULTIPATH

Multipath effects in underwater sound transmission result from the fact that, in general, sound energy from a source may reach a receiver by many different ray paths (see figure below). These paths result from reflection of the sound rays at the surface and bottom of the ocean as well as varying refractive effects over different paths. Because of the different arrival times at the receiver associated with the paths, the result is often called the "image effect."



NOTE: Rays may be curved due to refraction effects.

NOISE

Noise normally means any spurious or undesired disturbances that tend to obscure or mask the signal to be transmitted or received. These disturbances can originate from a large number of sources, both internal and external to the sonar system itself (see FLOW NOISE, SELF NOISE, AMBIENT NOISE, etc.).

A. Colored - Colored is a descriptive modifier which, when applied to the word noise, implies a nonuniform distribution of the noise energy as a function of frequency. Normally the nonuniform distribution is specified over some given frequency range.

B. White - White is a descriptive modifier which, when applied to the word noise, implies a uniform distribution of the noise energy as a function of frequency. Normally, the uniform distribution is specified over some given frequency range.

C. Background Noise - Background noise is the total noise reaching the operator of a sonar system. It is there even when a signal is present. This definition can be applied to either an operator listening to sonar audio or to an operator viewing a video sonar display.

Background noise is classified according to its origin as (1) SELF NOISE, which originates in the vessel and sonar receiving system, and

(2) AMBIENT NOISE, which is that noise existing in the ocean medium in the absence of any man-made sources.

D. Self Noise - Self noise is the noise originating in the listening vessel and the receiving system. Self noise has three sources: (1) the receiving system itself generates thermal noise, tube noise, microphonics, etc., or it may pick up man-made interference, (2) the hydrophones, in moving through the water, cause acoustical disturbances, and (3) the vessel serving as platform for the receiving system generates noise both by its engines (and possibly other equipment) and by its motion through the water. This noise is referred to as FLOW NOISE.

E. Ambient Noise - Ambient noise is the component of BACKGROUND NOISE in the sea caused by natural phenomena such as surf, wave action, rain, or impact masses of water, escape of entrapped air bubbles, and biological noise caused by fish and other sea life. It is the limiting noise field existing when all man-made noise is eliminated.

To a first approximation, ambient noise may be assumed to be ISOTROPIC, though recent measurements indicate this assumption is not valid for detailed considerations. Ambient sea noise spectra have been measured by Wenz and Knudsen. In the "Knudsen curves," measured ambient sea noise spectrum level is plotted versus frequency, with sea state as parameter. Over the frequency interval 2-20 kHz these curves show a slope of approximately -6 dB per octave.

F. Flow Noise - Flow noise is the result of the acoustic disturbances that arise, directly or indirectly, from the motion of a vessel through the medium. Direct flow noise is due in part to motion-induced turbulence along the hull of the vessel, and in part to a little-understood mechanism that is present even in the absence of turbulence, i.e., when the flow is laminar. Indirect, or induced, flow noise is due to disturbances caused by the flow, which in turn produce acoustic disturbances, such as flow-induced vibrations in the hull, which in turn, couple to the water and generate noise.

G. Gaussian Noise - Gaussian noise is noise for which the probability distribution of its instantaneous amplitude is Gaussian. It is frequently convenient, for purposes of analysis, to assume that the noise in a particular case is Gaussian, particularly when the exact distribution is not known. However, while many types of noise are in fact Gaussian or approximately so, there are also many cases where this assumption is poor, e.g., impulse noise.

H. Radiated Noise - Radiated noise is that noise emanating from a source that is detected or sensed by a remote sonar. Radiated noise from a target is of vital importance to passive sonar, and is of concern to submarine personnel in that their own detectability is a function of the level of their own radiated noise. Under these conditions the terms "noise" and "signal" are synonymous.

I. Isotropic Noise Field - An isotropic noise field is a noise field in which the cross-power density

spectrum of two omnidirectional receivers is independent of the location of the receivers and their orientation. However, the cross-power density spectrum is dependent on the physical separation of the two receivers. This field is most generally referred to as a spherical isotropic noise field.

For the spherical isotropic noise field, the receivers are assumed to be surrounded by an infinitely large number of independent noise sources. These noise sources are assumed to be uniformly distributed over the surface of a sphere of infinite radius. The noise radiated by these sources has identical power density spectra.

A two-dimensional isotropic noise field is referred to as a cylindrical isotropic noise field.

J. Anisotropic Noise Field - An anisotropic noise field is a noise field in which the cross-power density spectrum between two omnidirectional receivers is dependent on the receiver pair locations, orientation, and their physical separation. Generally, this type of noise field is not easily described by an arbitrary arrangement of farfield radiating sources as in the isotropic case.

However, an anisotropic noise field may be modeled by considering a composition of both nearfield sources and farfield sources. A simpler version of anisotropic noise field could be modeled by receivers surrounded by an infinitely large number of independent noise sources. These noise sources

are assumed to be uniformly distributed over the surface of an infinite radius cylinder.

In this simple version of an anisotropic noise field, the noise radiated by these sources need not have the same power density spectra. In the case of identical radiating noise sources, the anisotropic effects are introduced by adjusting the total noise power radiated by each source.

NORMAL MODE THEORY

The normal mode theory is a theoretical approach to describing acoustic propagation similar to that provided by RAY PATH theory. In normal mode theory, the sound field is described by an infinite set of characteristic functions, each of which is a solution of the wave equation.

For example, the one-dimensional wave equation

$$\frac{\partial^2 \psi}{\partial t^2} = c^2 \frac{\partial^2 \psi}{\partial x^2}$$

has the normal-mode solution:

$$x = \sum_{n=1}^{\infty} A_n \begin{matrix} \sin \\ \cos \end{matrix} (knx) e^{i\omega_n t} .$$

The term "normal" arises from the fact that all the modes (solutions) are orthogonal (normal) to each other, i.e.,

$$\int_0^{2\pi} \sin mx \sin nx \, dx = \delta_{mn}$$

$$\delta_{mn} = \begin{cases} 0 & m \neq n \\ 1 & m = n \end{cases}$$

$$\int_0^{2\pi} \sin mx \cos nx \, dx = 0 .$$

The normal mode solution is a very complicated mathematical function little used until the digital computer became

available to carry out the computations. Under certain conditions the normal mode theory and ray path theory provide nearly identical results, while under other conditions, e.g., SHALLOW WATER, very long ranges, "shallow zone" effects, caustics, etc., the normal mode theory provides a much more realistic solution.

NULL STEERING

Null steering is a beam forming technique which is used to discriminate against known NOISE sources. The direction of incidence on the array of many noise components is known; for example, screw noise, both direct and, in the submarine case, surface reflected. Just as individual transducer responses can be summed in a prescribed fashion to form a maximum array response in a desired direction for target DETECTION, the effect on array performance of known NOISE sources can be reduced by forming nulls or minimizing array response in the direction of those NOISE components.

While a PROCESSING GAIN can be achieved with null steering, any target on a bearing near the direction of the null will have a degraded signal level.

PARAMETER ESTIMATION

Parameter estimation refers to the statistical techniques of estimating the values of significant signal parameters in active and passive sonar, to aid in target TRACKING and CLASSIFICATIONS. Optimum methods of parameter estimation depend on the characteristics of the signal and the background interference in much the same way as optimum DETECTION strategies.

Examples of signal parameters whose values would be estimated are: (1) arrival time of the echo from a pulse in active sonar, to accurately determine the time interval between the transmission of the pulse and the reception of its echo; and (2) frequency of the received echo, to determine the velocity component of a moving target in the direction of the sonar.

PING-TO-PING INTEGRATION

Ping-to-ping integration is used in active sonar detection using multipulse transmission to enhance detection probability. Historically it has been accomplished by using multi-ping displays in which horizontal traces with amplitude brightening (of fluorescent screens) or darkening (of paper stylus recorders) is available. Each horizontal line on the display represents the amplitude variation of the received waveform over a ping cycle. When n such traces have been written adjacent to each other on the display, the presence of a target is indicated by a stripe on the display across the sequence of horizontal lines. The presence of the stripe is easier to detect than a darkened region on a single horizontal trace of one ping cycle. It has been found experimentally that displays with n traces provide an improvement in DETECTION capability equivalent to an increase in SIGNAL-TO-NOISE RATIO of 5 log n, or greater, measured at the processor input, relative to a single trace display. This is an experimental result based upon approximately equal FALSE ALARM rates and n values between n = 1 and n = 25.

PROBABILITY

Probability is a concept used to express the fraction of time an event will occur in a long series of trials. The event in question may be anything such as drawing a pair of playing cards out of a deck, ascertaining that a target is present in a specified location, or evaluating whether a weapon will be effective. Inherent in the probability statement is the concept of a large number of trials or occasions in which the event may occur. If N is the number of trials, and E is an event that happens n_N times, then the probability of event E, denoted Pr(E), is

$$Pr(E) = \lim_{n \to \infty} (n_N / N).$$

It should be emphasized that probability makes no statement about the outcome of a particular trial--just the long term aspects.

One may also define the useful concept of conditional probability. For example, one might be interested in the probability of the event of a successful weapons attack given that the event of a submarine DETECTION and classification has occurred.

PROBABILITY DENSITY FUNCTION

The probability density function, $p(x)$, gives the probability

$$p(x) = Pr(x \leq y \leq x + dx)$$

where dx is an infinitesimal increment. If the PROBABILITY DISTRIBUTION has a derivative at x, then

$$p(x) = \frac{df(x)}{dx} \quad .$$

PROBABILITY DISTRIBUTION FUNCTION

The distribution function of a RANDOM VARIABLE, x, is the function

$$F_x(y) = Pr(x \leq y)$$

for any number $-\infty \leq y \leq \infty$. The distribution summarizes probability information about events. Some of its properties are

1. $F(-\infty) = 0$; $F(\infty) = 1$

2. F is non-decreasing function of $F(y_1) \leq F(y_2)$ if $y_1 \leq y_2$.

If the PROBABILITY DENSITY FUNCTION, p(x), is continuous, then

$$F_x(y) = \int_{-\infty}^{y} p(x)dx.$$

PROBABILITY OF DETECTION AND FALSE ALARM

The probability of detection is the probability that a sonar operator will make a correct decision regarding presence of a target in a noise background. The probability of false alarm is the probability that a sonar operator will make an incorrect decision regarding the presence of a target in a noise background when only the noise background is present. These probabilities depend on the SIGNAL and NOISE FIELDS present, the sonar system signal processing, and the THRESHOLD level set for the values of the sonar output variables upon which a DETECTION decision is to be based.

PROCESSING GAIN

The processing gain of a receiver is defined as the ratio of the SIGNAL-TO-NOISE RATIO at the output of the receiver to the SIGNAL-TO-NOISE RATIO at the input of the receiver. This ratio is usually expressed in DECIBELS. Although useful for comparing receivers, processing gain describes the performance of only a part of the input to the decision chain and has only limited value as a working concept for the overall DETECTION system.

PROPAGATION LOSS

Propagation or transmission loss is a term which quantifies the decrease in acoustic intensity below the source level.

A major factor affecting propagation loss is spreading. As the waves spread out from a source the surface across which the energy is being propagated becomes larger and larger. Near a source the spreading is spherical and the intensity is proportional to the inverse square of the distance. At larger distances the spreading is affected by refraction, which causes the rays to curve. Refraction is capable of producing drastic changes in the spreading loss, including focusing effects and shadow zones.

Another factor affecting propagation loss is attenuation. Attenuation refers to the loss of energy caused by absorption and scattering.

Absorption losses in the sea may be divided into three basic types, viscous losses, heat conduction losses and losses associated with molecular exchanges of energy. The viscous losses result from relative motion occurring between various portions of the medium during the compressions and expansions that accompany transmission of a sound wave. Heat losses result from a tendency for heat to be conducted from regions of compression where the temperature is lowered. In the process of this heat transfer there is a tendency toward pressure equalization, which reduces the amplitude of a wave as it is propagated through the medium. The dissipation of acoustic energy associated with changes in the molecular structure of the medium results from the finite time required for these changes to occur. For example, when the period of the acoustic cycle is comparable with the time

required for a portion of the compressional energy of the fluid to be converted into internal energy of molecular vibration, then correspondingly during the expansion cycle some of this energy will be delayed in its restoration so as to be returned to the fluid during a time of expansion. Such a delay will result in a tendency toward pressure equalization and an attendant reduction in pressure amplitude of the wave.

Scattering refers to the process whereby objects in the water (bubbles, marine life, etc.) cause a portion of the energy in a wave to be deflected in various directions in the form of incoherent radiation and thereby rendered useless. Scattering can also be caused by the surface and the bottom of the ocean. Usually, however, the greater part of the energy scattered by the surface and bottom is scattered in the specular direction (that direction for which the angle of reflection equals the angle of incidence). Scattering is also caused by the thermal macrostructure of the ocean.

Q - QUALITY FACTOR

Q or quality factor is a measure of the sharpness of resonance or frequency selectivity of a tuned electrical or mechanical system with a single degree of freedom. For an active sonar transducer, the significance of the mechanical Q is in relation to the "rise time" of the transducer, or the time required for the maximum acoustical output to be achieved after a sinusoidal electrical input signal is started. A transducer with a high Q in general has a longer rise time. For a passive receiving sonar intended for broadband operation, a low value of Q is necessary to achieve a nearly uniform response over the frequency band.

Q may be defined explicitly in a variety of ways, but it is most fundamentally significant as an expression of the ratio of the peak energy stored during a cycle to the total energy dissipated in a period. If f_o is the resonant frequency, and f_1, f_2, the "half-power" frequencies, i.e., the frequencies above and below resonance where the output power is one half the peak value at resonance, Q may be expressed as:

$$Q = \frac{f_o}{f_1 - f_2} .$$

QUENCHING

Quenching is the introduction of occluded air, bubbles or other discontinuities into the transmission path of acoustic energy. The resulting PROPAGATION LOSS due to attenuation is abnormally high. Quenching usually occurs where there is considerable water turbulence, as in SHALLOW WATER or near a shore with strong currents, with vessels having a strong wake, in high sea states when ship motion causes the transducer to break the surface, or when high ship speeds cause surface bubbles to sweep down around the transducer. In severe cases of quenching, where the transducer pulses into an air load, as may occur in rough seas and/or high speeds, the resulting impedance mismatch may cause damage to the transducer elements.

RANGING

Ranging is the process of determining the physical separation between a sensor and a target. Ranging can be accomplished by either active or passive means. Active ranging is achieved by measuring the travel time of a sonar pulse and dividing this by the average speed of sound along the probable path of acoustic propagation. Passive ranging is accomplished by employing azimuthal and/or vertical signal arrival angle information or by correlation techniques.

Passive ranging techniques employing azimuthal data consist of both manual and automatic ranging techniques. Each of these techniques (known as bearings-only techniques) combines measures of azimuthal signal arrival angle over a period of time with assumptions about target motion parameters to yield estimates of target range. Typical manual techniques are Ekelund, Spiess, Strip Plot, Four Bearing and Relative Motion Plot. Automated techniques are MK-7 Analyzer, MK-112 Analyzer, MK-51 Analyzer (Churn Routines) and MATE Routines.

Ranging routines based on depth plane data utilize vertical signal arrival angle with assumptions concerning target motion and the environment to predict target range. Manual techniques (Depth/Elevation Angle Ray Trace Ranging, Propagation Loss Ranging) and automated techniques (MK-51 Analyzer with Depth/Elevation Angle Input) are available.

Correlation techniques perform cross correlations between signal received at several sensors to determine signal arrival time differences at the sensors. These arrival time differences are then extrapolated to yield target range estimates.

RANDOM VARIABLE

Probability statements concern events, such as whether a coin shows heads or tails. It is often convenient to relate these events to numbers. The function, X, which accomplishes this is called a RANDOM VARIABLE. For instance, for a toss of a coin,

$$X(T) = 0 \text{ if tails is realized,}$$

$$X(H) = 1 \text{ if heads is realized.}$$

The value of the random variable, 0 or 1 in the above sample, is often loosely referred to as a random variable.

RAY PATHS

A ray path is the route along which sound energy is propagated under the influence of refraction as the sound velocity varies from point to point.

RAY THEORY provides a more convenient but somewhat less rigorous analytical tool than the wave equation (NORMAL MODE) for the calculation of sound intensity at a point in the ocean. Practical methods for determining the ray path include both graphical plotting devices and numerical algorithms suitable for high speed computer use. RAY THEORY provides a useful method for the theoretical determination of TRANSMISSION LOSS in the ocean, and hence ray path analysis is used extensively in performance analyses of sonar systems.

RAY THEORY

Ray theory is a mathematical construct useful in describing phenomena associated with the propagation of sound in the sea. Ray theory, which is analogous to geometrical optics, is intuitively appealing since it presents a picture of propagation paths through the sea in the form of ray diagrams. The fundamental postulates of ray theory are (1) the existence of wavefronts which are surfaces of constant phase and (2) the concept of rays which are normal to the wavefronts and give the direction of propagation of the acoustic energy through the medium. Ray theory is particularly useful in describing PROPAGATION LOSS and REVERBERATION levels for modern active sonars operating in the direct path or BOTTOM BOUNCE propagation modes. The limitations of ray theory are that it does not provide realistic solutions under conditions where the radius of curvature of the rays or the pressure amplitude changes appreciably over the distance of one wavelength. Hence, wave theory is not generally applicable at low frequencies or for predicting the intensity of sound in shadow zones or CAUSTICS.

The fundamental equation of ray theory is called the eikonal equation which is a transformation of the Helmholtz equation into a first-order time-independent partial differential equation. In effecting the transformation it is necessary to invoke the assumption which defines the limitations of ray theory as pointed out in the previous paragraph. A detailed discussion of the mathematical aspects of ray theory can be found in <u>Introduction to the Theory of Sound Transmission</u>, by C. B. Officer, McGraw-Hill, 1958, pp. 36-61 or <u>Physics of Sound in the Sea, Part I</u>: Transmission, Chapter 3, U. S. Dept. of Commerce, Office of Technical Services.

RECEIVER OPERATING CHARACTERISTIC (ROC)

A receiver operating characteristic or curve is a plot of the PROBABILITY OF DETECTION versus the PROBABILITY OF A FALSE ALARM for a fixed value of the SIGNAL-TO-NOISE RATIO at the input of the receiver.

The PROBABILITY OF DETECTION, P(D), and the PROBABILITY OF A FALSE ALARM, P(FA), can be expressed in the form

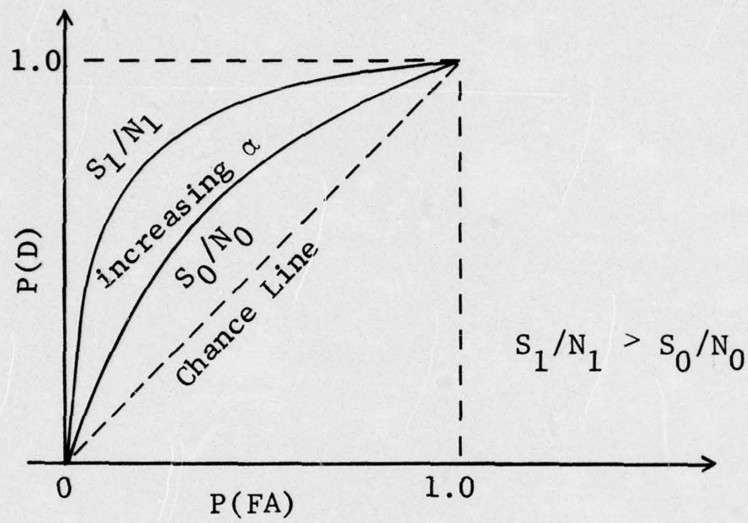
$$P(D) = \int_{\alpha}^{\infty} \rho_{s+n}(y, S/N)\, dy,$$

$$P(FA) = \int_{\alpha}^{\infty} \rho_{n}(y, S/N)\, dy,$$

where α is the detection threshold setting and S/N is the signal-to-noise ratio at the input of the receiver. The RANDOM VARIABLE y is the LIKELIHOOD RATIO and $\rho_{s+n}(y,S/N)$ is the PROBABILITY DENSITY FUNCTION for y under the hypothesis that the waveform presented to the receiver contains signal and noise while $\rho_n(y,S/N)$ is the probability density function for y under the hypothesis that the waveform presented to the receiver contains noise only. (For a derivation of the above equation see PROBABILITIES OF DETECTION and FALSE ALARM.)

Now assume that the SIGNAL-TO-NOISE RATIO is held fixed, say at the value S_0/N_0. Then from the above equations it is seen that P(D) and P(FA) are functions only of α which is the lower limit of integration. Thus for each value of α, the indicated integration can be performed and a value of P(D) and P(FA) obtained corresponding to that value of α, it being assumed that

ρ_{s+n} and ρ_n are known functions of y and S_0/N_0. In this manner, a plot of P(D) versus P(FA) is obtained as α is allowed to vary. Such a curve is shown below.



RECEIVER OPERATING CHARACTERISTIC CURVES
FOR FIXED VALUES OF S/N

This curve is one member of a family of curves called receiver operating characteristics (ROC). The other members of the family are generated by allowing S/N to vary. For example consider a second value of the signal-to-noise ratio, say $S_1/N_1 > S_0/N_0$. A comparison of the receiver operating characteristic for the signal-to-noise value S_1/N_1 and the receiver operating characteristic for the value S_0/N_0 is seen in the figure.

RECEIVER OPERATING POINT

The receiver operating point is synonymous with the level at which the THRESHOLD is set, which establishes the weakest signal that is detectable. If the receiver output exceeds the threshold, a signal is assumed (for binary, non-sequential processors) to be present. If the threshold is not exceeded, any signals which are present will not be detected. Lowering the threshold level so that weaker signals will be detected increases the liklihood that noise alone will exceed the threshold, giving rise to more FALSE ALARMS. The selection of the proper threshold level is a compromise that depends on the improtance of making a mistake by (1) failing to recognize a signal that is present (probability of a miss) or by (2) falsely indicating the presence of a signal when none exists (PROBABILITY OF A FALSE ALARM). The establishment of the threshold level is based on a specified FALSE ALARM rate and, in turn, determines the PROBABILITY OF DETECTION and the PROBABILITY OF FALSE ALARM.

RECOGNITION DIFFERENTIAL

The recognition differential (RD) for a sonar system is generally an experimental quantity obtained by permitting human subjects to attempt to detect signals in a background noise under constant SIGNAL-TO-NOISE RATIO (SNR) conditions for some period of time. That SNR that results in 0.5 probability of detection or at which the target is detected 50% of the time is termed the RD. The RD for a sonar system can be estimated by assuming that the operator acts as an integrator.

In comparing recognition differentials for various sonar systems, it is important to first ascertain that the factors associated with RD (i.e., observation time, SNR definition, noise clutter density, probability of DETECTION, FALSE ALARM probability, and measurement techniques or computational methods) are equivalent. It should also be remembered that RD applies to static conditions only, and realistic sonar detection situations often involve important dynamic phenomena that cannot be examined by the RD approach to performance modeling.

REVERBERATION

Reverberation, in active sonar, is acoustic energy returned to the receiver by reflection from within the volume of the medium and its boundaries. Volume REVERBERATION results from myriad small scatterers and INHOMOGENEITIES present in the ocean, each of which returns a small echo to the sonar receiver. Boundary REVERBERATION consists of scattering from the ocean surface and bottom; hence boundary reverberation is further classified as surface and bottom reverberation. Reverberation differs from BACKGROUND NOISE in that its frequency content is determined by the frequency characteristics of the transmitted pulse and by DOPPLER effects. In addition, reverberation decays with time, whereas the background noise does not.

The character of reverberation as a function of time after transmission depends upon a number of factors assocaiated with the sonar and the transmission medium. These factors are listed below:

a. Source level

b. Pulse length

c. Transmit beam pattern

d. Receiver beam pattern

e. Depression/elevation angles of the beam axes

f. Sonar depth

g. Water depth

h. Volume scattering strength

i. Surface scattering strength

j. Bottom scattering strength

Volume scattering strength depends primarily upon frequency and the density and average size of biological scatters in the DEEP SCATTERING LAYER. Surface scattering strength depends on frequency, wind speed and angle of incidence of acoustic energy. Bottom scattering strength is basically dependent upon frequency, angle of incidence, and bottom type. A good disucssion of reverberation is given by R. J. Urich in Principles of Underwater Sound for Engineers, Chapter 8, McGraw-Hill, 1967.

SAMPLING THEOREM

The sampling theorem prescribes the number of and interval between samples of a waveform of signal which must be taken to describe that waveform. It states that if U(t) is an arbitrary waveform whose spectrum (Fourier transform) vanishes identically outside the finite frequency interval (-W,W) then U(t) is completely determined by its values at equally spaced time intervals of width $\frac{1}{2W}$, where W is the bandwidth of the waveform. Mathematically this result may be expressed as a series representation of U(t):

$$U(t) = \sum_{k=-\infty}^{\infty} U\left(\frac{k}{2W}\right) \frac{\sin 2\pi W\ (t - k/2W)}{2\pi W\ (t - k/2W)}$$

If U(t) is in addition periodic in time, only a finite number of samples is needed to completely determine U(t).

The significance of the sampling theory for sonar operations is in the necessary sampling rate and data rate which a detector must have to adequately recover a random waveform from an input signal.

SEARCH RATE

Search rate indicates the total area which can be searched per unit time, and is most frequently measured in square miles per hour. Search rate further implies a search effectiveness in terms of the probability that a target within the searched area will be detected.

The searching sensor, sonar, provides the searcher with an effective lateral range, i.e., the greatest range of the closest-point-of-approach of a target TRACK at which the target will be detected with a specified PROBABILITY OF DETECTION. Assuming an omnidirectional sensor, the sweep width for the searcher will be twice the lateral range, and the search rate will be the product of the sweep width and the searchers's speed. Thus a 10 knot search vessel with an effective sonar DETECTION range of 6 Kyd (3 n.mi.) will achieve an effective search rate of 2 x 3 x 10 = 60 sq. n. mi/hour, with, for example, A PROBABILITY OF DETECTION (P_D) of 0.50 for any target in the searched area.

Increasing the searcher's speed increases search rate, but, with sonar, decreases the lateral range due to increased self-noise. Thus the search rate must be maximized by selection of optimum ship speed relative to sonar performance.

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A GLOSSARY OF SONAR TECHNOLOGY TERMS. (U)

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SHADING

Shading is a SPATIAL PROCESSING technique by which the phases or amplitude responses of transducers in an array are varied to achieve a desired spatial distribution of radiated energy or receiver beamformer output. Amplitude shading is much more extensively employed in underwater sound than phase shading and is accomplished by adjusting the amplitude of the transducer response. Phase shading corresponds to the introduction of phase delays or advances of the spacing of array elements to achieve the desired phase variation.

Shading is usually employed to control some aspect of a DIRECTIVITY pattern such as side lobe level or main beam width. As a general rule, reduction in side lobe level is accompanied by increased main beam width and vice versa. One exception to this is the superdirective array in which narrow beams are obtained with small arrays at the cost of low sensitivity and high side lobes.

SHALLOW WATER OPERATION

In sonar technology there is no single, accepted definition of shallow water. Traditionally, the 100-fathom curve, such as found along the continential shelf, has been considered to mark the transition point between shallow and deep water. More recently, the criterion for shallow water has been expressed in terms of the ratio of water depth to sonar wavelength. If the water is only a few wavelengths in depth then it will likely be considered shallow water by the analyst concerned with modeling transmission loss. In general, water can be considered as shallow if the ocean bottom has a significant influence on the portion of the sound field that is of interest.

Ocean bottom characteristics are of primary importance in shallow water operation. For active sonars in SHALLOW WATER, high-level bottom REVERBERATION is usually the limiting background for target DETECTION. Also, shallow water PROPAGATION LOSS usually depends strongly on bottom properties due to the relatively large number of bottom-reflected MULTIPATHS involved in sound propagation between sonar and target.

SIGNAL FIELD

A signal field is the spatial distribution, in the ocean medium, of acoustic energy from a radiating source such as a sonar transducer or transducer array, or a radiating target. The signal field is divided into two parts: the near field (Fresnel Zone), and the far field (Fraunhofer Zone).

The acoustic intensity pattern in the near field contains a complex arrangement of peaks and nulls created by interference. The acoustic intensity at any point in the field is the result of summing the acoustic energy arriving at that point from the entire radiating surface. Since the path length to any point in the sound field varies with position on the radiating surface, the summed energy at the point contains components of many phases. Peaks and nulls in the pattern result from constructive and destructive interference, respectively, among the arrivals. In the case of some high-powered active transducers, these peaks and nulls can be so severe as to cause CAVITATION in the near field.

In the far field, by contract, where distance from the source is very large compared to the dimensions of the radiating source, there are no peaks or nulls, because the wave fronts are flatter and have fewer intersections. The acoustic intensity in the far field varies as a function of frequency, water condition (temperature, depth, salinity, particulate matter, etc.), surface and bottom conditions (surface vaves, bottom roughness and hardness) and range.

The distance from the radiating source which marks the transition from the Fresnel zone to the far field depends on the wavelength of the emitted sound and the characteristics of the

radiating source. Generally, the transition from near field to far field is assumed to occur about 15 wave lengths from the source, for active sonars. In most sonar applications, the far field signal field is the most significant.

SIGNAL-TO-NOISE RATIO

Signal-to-noise ratio is the ratio of signal power to noise power occurring at a specified point. The power quantities are expressed in terms of acoustic intensity or acoustic intensity per unit frequency band, when dealing with acoustic waves in the water; in terms of electric power or electric power per unit band when dealing with electric waves in a sonar system component.

Since instantaneous values of both signal and noise powers are usually RANDOM VARIABLES, many definitions of signal-to-noise ratio are possible, depending on the specific values, e.g., peak, average, etc., of signal power and noise power that are used. The particular definition that is used will depend on the purpose in mind. The signal-to-noise ratio of the ultimate response of a sonar system is a significant figure-of-merit indicator in the design and operation of sonar equipment. However, caution must be taken in comparing or using values of signal-to-noise ratio from different sources, to be sure they are comparable or have compatible bases for comparison.

SIGNATURE

The passive signature of a target is the set of characteristics of the acoustic SIGNAL FIELD radiated from the target. The signature of a submarine target is characterized by a typical broadband radiation plus a set of line components at discrete frequencies. The frequencies of the observed line components vary from time to time as the target changes speed, bearing and the machinery in use.

A PRIORI knowledge of some of the target signature characteristics is useful to improve passive TIME PROCESSING, and hence to improve the DETECTION performance of the sonar system. (ADAPTIVE processing may be employed to predict the target signature structure.)

SIMULATION

Simulation is a technique for conducting experiments with mathematical and logical models that describe the behavior of systems (or components thereof). The simulation models may be constructed to produce experimental outcomes which are of three general types:

a. DETERMINISTIC: the experiment possesses unique outcomes for a given set of inputs;

b. STOCHASTIC: functional relationships depend on chance parameters and the experimental outcomes can be predicted only in a probabilistic sense;

c. EXPECTED VALUE: expected values are assigned to chance parameters in the models.

In simulation model construction, the system to be simulated is divided in terms OF PROBABILITY distributions, for each of the possible states of the element and its inputs. The models of these elements are then joined in the proper sequence and the interrelationships between each element built into the simulation model. Input data are generated and the system model's behavior recorded.

For example, the performance of a sonar signal processing device can be simulated in terms of the sequential response of filters, averagers, etc. to the respective input data. Similarly, the performance of a screen of ASW ships can be investigated by simulating the performance of each ship in the screen against a common threat.

SONAR

Sonar, an acronym for Sound Navigation and Ranging is the branch of applied acoustics that deals with the generation, emission, propagation and reception of acoustic energy underwater; and the utilization of underwater sound for navigation; object DETECTION, CLASSIFICATION, LOCALIZATION and TRACKING; and communication.

A. An active sonar system generates and emits acoustic energy specifically for the purpose of obtaining information concerning a distant object from the received and processed reflected sound energy. Advantages: gives acoustic range information; good versus a quiet target; speed information from DOPPLER SHIFT. Disadvantages: alerts target to searchers; range limited by two way propagation loss; signal strength depends on target aspect; REVERBERATION limits sound transmission.

B. A passive sonar system determines information about a distant object by receiving and processing acoustic energy which is generated and emitted by that object. Advantages: does not alert target; has only one way propagation loss; creates no reverberation; enhancies ability to classify targets. Disadvantages: depends upon cooperation of target; gives poor range information.

SOUND PRESSURE

The sound pressure at a point is the total instantaneous pressure at that point in the presence of a sound wave, minus the static pressure at that point.

a. EFFECTIVE SOUND PRESSURE (RMS SOUND PRESSURE). The effective sound pressure at a point is the root-mean-square value of the instantaneous sound pressure, over a time interval, at the point under consideration. In the case of periodic sound pressures, the interval must be an integral number of periods, or an interval that is long compared to a period. In the case of nonperiodic sound pressures, the interval should be long enough to make the value obtained essentially independent of small changes in the length of the interval. The term, effective pressure, is frequently shortened to sound pressure.

b. MAXIMUM SOUND PRESSURE. The maximum sound pressure for any given cycle of a periodic wave is the maximum absolute value of the instantaneous sound pressure occurring during that cycle. For a sinusoidal wave, this is also the pressure amplitude.

c. PEAK SOUND PRESSURE. The peak sound pressure for any specified time interval is the maximum absolute value of the instantaneous sound pressure in that interval. For a periodic wave, if the interval is one period, the peak and maximum sound pressures are identical.

d. STATIC PRESSURE. The static pressure at a point is the pressure that would exist at that point in the absence of sound waves.

SOURCE LEVEL

Source level is the term used to specify the acoustic intensity radiated by a sound projector. For underwater acoustic projectors it is defined as the intensity of the radiated sound in DECIBELS relative to the intensity of a plane wave having an acoustic pressure of either 1 microbar (1 dyne/cm^2) <u>or</u> 1 micropascal (10^{-5} dynes/cm^2), referred to a distance of 1 yard from the acoustic center of the projector in the direction of a traget. Note: (One microbar was the standard reference until 1971 and is the most common reference in current use. The micropascal was introduced as the new reference by NAVSHIPS Notice 9400, dated 12 January 1971. A gradual transition to the new unit is expected by requiring its use in all new NAVSHIPS contracts. A source level referenced to 1 microbar is converted to a reference of 1 micropascal by adding 100 dB.) Source level is related to the radiated acoustic power, P, by the equation

$$SL(dB//1\mu\ bar) = 71.5 + 10 \log P + DI_T,$$

where P is in watts and DI_T is the DIRECTIVITY INDEX of the projector in dB. The factor 71.5 is a conversion factor, relating the reference acoustic power, 1 watt, to the reference sound pressure, 1 dyne/cm^2. For source levels referred to 1 micropascal, this conversion factor is 171.5.

SPATIAL CORRELATION

Spatial correlation is the process of comparing signals received by sensors at different locations in space. Referring to the definition of temporal correlation, the cross-correlation function of two random variables, x_1 and x_2, is given by

$$\Phi_{12}(\tau) = \lim_{T\to\infty} \frac{1}{2T} \int_{-T}^{T} x_1(t)\, x_2(t + \tau)\, dt.$$

The additional complexity added in SPATIAL PROCESSING is that the time shift τ is, in some manner, affected by geometrical considerations. Consider two points in space, x_1 and x_2, and assume that there is some random process affecting both of them Now, in addition, assume that a plane wave is approaching them. Certainly the cross-correlation of the two RANDOM VARIABLES $x_1(t)$ and $x_2(t)$ will be a function of the angle that the plane wave makes with the line joining the two points. If that angle is zero, $\Phi_{12}(\tau)$ will have its maximum at $\tau = 0$. The largest value of τ, for the maximum of $\Phi_{12}(\tau)$, will occur when the wave is normal to the line joining x_1 and x_2.

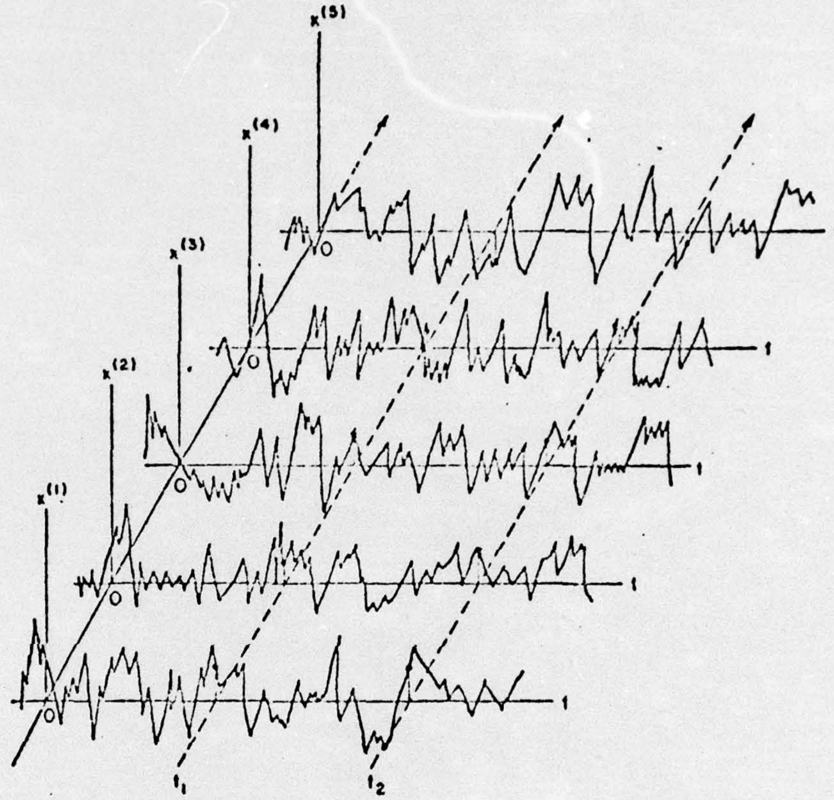
SPATIAL PROCESSING (BEAMFORMING)

Spatial processing and beamforming as applied to receiving transducer arrays may be defined generally as that part of signal processing which utilizes the spatial positions of the transducers or transducer elements to enhance output SIGNAL-TO-NOISE RATIO. Included in spatial processing are aperture selection, i.e., the choice of a particular group of transducers from the array to be used in forming a desired receiving beam; amplitude SHADING and time delays at the outputs of the various transducers used, and SPATIAL CORRELATION of the various transducer outputs to discriminate against NOISE.

STOCHASTIC PROCESSES

A stochastic process is a collection of RANDOM VARIABLES, X_t, all defined over the same probability space, S, of events and indexed by a parameter, t. The indexing parameter is usually time in sonar applications, and may be either discrete or continuous.

As an example of a stochastic process consider the following experiment. A transducer is held at a fixed depth in the ocean and a pulse is put into the water. The returning echo is recorded as a function of time. Imagine this experiment performed an infinite number of times. Let $X^{(i)}(t) = X(t,\xi_i)$ be the voltage recorded as a function of time for the returning signal on the ith trial. In this manner a whole family or ensemble of time functions are generated. An individual time function $X^{(i)}(t)$ is called a sample function. The figure shows a few of the infinite number of different possible time functions. Each sample function, i.e., each echo, is different due to random fluctuations in variables not under control of the experimenter. For example, the fluctuation due to a continually changing ocean surface which will be different from one pulse to the next, i.e., from one trial to the next.



SCHEMATIC REPRESENTATION OF A STOCHASTIC PROCESS X(t).

(Each $X^{(i)}(t)$ is a sample function of the ensemble.)

SURFACE DUCT

Under certain weather conditions a layer of isothermal water will extend from the ocean surface to a depth typically on the order of 50 to 400 feet. Deep layers are most often found in the winter. In the summer the layer is usually shallow or non-existent. This isothermal layer is called the surface duct. Within the duct the speed of sound increases with depth due to the increase in hydrostatic pressure. The net result is a positive sound velocity gradient from the ocean surface to layer depth.

Sound energy emitted from a source in the duct along ray paths near the horizontal tends to be trapped in the duct due to the upward bending of the rays (refraction) induced by the positive gradient. This refracted sound energy continues to propagate outward in range in the duct through repeated surface reflections.

For certain favorable combinations of layer depth, frequency, and surface roughness the propagation loss in the duct is approximately characterized by cylindrical rather than spherical spreading. Submarines seeking to avoid DETECTION by near surface sonars will generally travel well below the surface duct.

TARGET STRENGTH

Target strength in active sonar is the ratio of reflected acoustic intensity at a point one yard from the effective reradiation center of the target, to the incident acoustic intensity. Target strength is a measure of the reflectivity of the target to acoustic energy. It is a function of the type, size, shape, and nature of the target, as well as of its orientation or aspect. Measured target strengths of beam aspect submarines lie in the 20-25 dB range; for bow-stern aspect, the range is 5-10 dB; while for oblique aspect target strength ranges from 15 to 25 dB.

THRESHOLD

A threshold level associated with a sonar receiver output variable is a value of the variable that is chosen A PRIORI and is used in making a DETECTION decision. The decision concerning the presence of a signal is made on the basis of whether or not the threshold is exceeded by the values of this variable. The threshold value chosen usually depends on such a priori information as the desired noise clutter density.

TIME-BANDWIDTH (BT) PRODUCT

Time-bandwidth (BT) product of a sonar detector is a dimensionless parameter equal to the product of the averaging time used in the processor and the frequency bandwidth of the acoustic input to the processor. The output statistics of the processor are a function of the time-bandwidth product and the input statistics; these in turn determine significant detector performance parameters such as PROBABILITY that signal plus noise or noise alone exceeds some threshold value. For small time-bandwidth products (BT = 1), the processor output statistics are nearly Rayleigh. As the time-bandwidth product is increased, the output statistics approach a GAUSSIAN form. For many input statistics and processor types, a BT product of 50 produces GAUSSIAN statistics for all but extremely small (on the order of 10^{-3} or less) probabilities.

TIME COMPRESSION

Time compression is a technique whereby a large number of data samples are stored as they arrive at their real-time rate, the older samples being replaced by new samples as they arrive. Between replacements the entire stored data sequence is available, for spectrum analysis, correlation, or other processes, if all of the stored data samples can be accessed and used in analysis or other processes before the next real-time arrival of a new data sample. If T is the real-time extent of the stored data and τ is interval between real-time samples, it is necessary to be able to access and use T seconds worth of data in τ seconds. The access and use of the data is accomplished T/τ times as fast as real time. The time compression ratio is referred to as T/τ. There is a corresponding increase in effective frequency of the time-compressed series of samples. These effects are illustrated in the definitions of DELTIC and DELTIC CORRELATORS, etc.

See also: DELTIC, DELTIC CORRELATOR

TIME PROCESSING

Time processing is that portion of both active and passive signal processing which usually occurs after SPATIAL PROCESSING (BEAMFORMING). The objective of time processing is to increase the SIGNAL-TO-NOISE RATIO of input signals (imbedded in noise) to make them more detectable. To do this, time processors are designed to exploit some aspect of the temporal differences which exist between signal-plus-noise waveforms and noise-alone waveforms.

In general, time processors are composed of both linear and nonlinear elements, and alter both the frequency content and statistics of the input waveforms. Time processing generally is considered to be operation which take place in the time domain, however, each time domain operation on a waveform may be viewed also as an equivalent operation in the frequency domain. For example, time domain averaging is equivalent to low-pass filtering in the frequency domain.

Time processors typically are grouped into either incoherent or coherent processors. Incoherent processors use little or no A PRIORI information describing the temporal structure of the processed waveform, while coherent processors are designed to utilize specific knowledge of waveform characteristics.

Incoherent time processors include the detector-averager which provides as an output a value proportional to the power present in the input waveform, and the spatial cross-correlator which detects similarities between samples of the signal imbedded in two differing noise waveforms.

An example of a coherent processor is the replica correlator which computes the cross-correlation between the input signal-plus-noise waveform and a stored replica of the transmitted signal.

TRACKING

The term "tracking" in sonar has two primary applications which are seldom referred to explicitly in the literature, or in general conversation. One is of a geometrical nature and the other is involved in signal processing.

"Geometrical tracking" in both active and passive sonar is the determination of the spatial position of a target as a function of time. Tracking accuracy is dependent on range and bearing accuracies of the sonar system as well as the acoustic environment. For navigational purposes the position of the target must be determined in a two-dimensional coordinate system fixed in space, while for fire control a three-dimensional position determination must be made. This concept of tracking is essentially a series of sequential LOCALIZATIONs from which the target track and speed can be derived and the future target position can be predicted.

"Signal processing tracking" is the process of keeping a positive identification on a signal as time progresses. Typical signal parameters requiring this type of tracking may be amplitude, frequency, phase or a combination of these measurements.

TRANSFER FUNCTION

A transfer function of a linear system is the ratio of the value of some physical variable at the system output to the value of a physical variable at the system input. For a sonar transducer, transfer impedance (ratio of output voltage to input current) is an example of a transfer function. In a linear signal processing system, PROCESSING GAIN is an important transfer function. The transfer function of a linear system may be defined more generally in terms of either the Fourier Transform or the Laplace Transform.

TRANSIENT

A transient in SONAR technology is a non-periodic, usually impulse-like, radiation of acoustic energy. Transients are normally of interest only in passive sonar operation where the receiving bandwidth is wide enough to pass detectable transient signals. Transients are predominantly produced by man-made sources such as operating submarine bow or stern planes or, rudder, or dropping a wrench on a steel deck, or the sound of a door closing, etc.; these transients are extremely useful in the CLASSIFICATION process.

TRANSIENT RESPONSE

A dynamical system is said to be in the steady state when the variables describing its behavior are either invariant with time, or are (sections of) periodic functions of time. A dynamical system is said to be in the TRANSIENT (or unsteady) state when it is not in the steady state. From a physical point of view it may be said that a TRANSIENT state exists in a physical system while the energy conditions of one steady state are being changed to those of a second steady state.

Certain TRANSIENT phenomena accompanying the redistribution of energy in a physical system are often wholly undesirable yet inescapable accompaniments of such a change. For example, when the driving force of a transducer element is suddenly started or stopped an undesirable TRANSIENT vibration results.

In contrast to these uncontrolled and unwanted transients there are desirable and controlled transients which are put to many important uses. The basis for self regualtion in all self-controlled systems is TRANSIENT in behavior. Examples are automatic systems for the control of voltage, frequency and sound-volume level. Thus dynamical systems need to be studied in the TRANSIENT state -- some for the purpose of minimizing unwanted TRANSIENT effects, other than for the purpose of controlling and employing desirable effects.